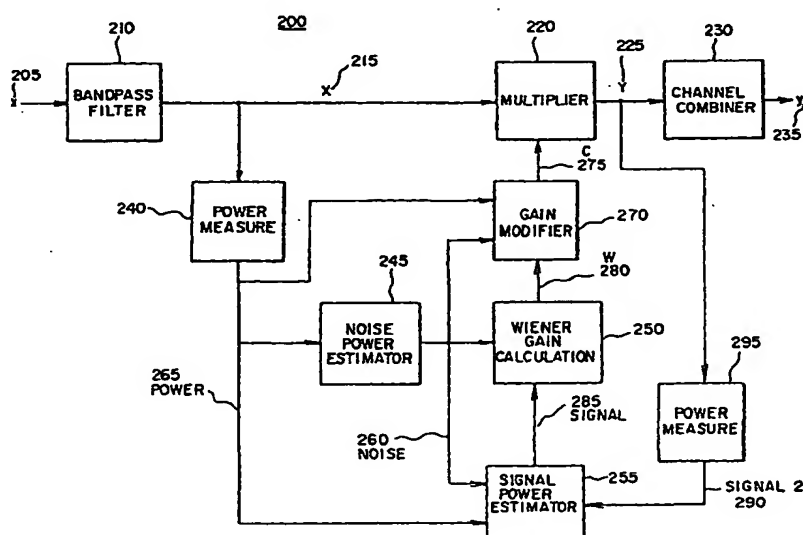




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(54) Title: A METHOD AND SYSTEM FOR UPDATING NOISE ESTIMATES DURING PAUSES IN AN INFORMATION SIGNAL



(57) Abstract

The invention relates to an improved adaptive spectral estimator for estimating the spectral components in a signal (205) containing both an information signal, such as speech, and noise. A method and system provide for generating noise estimates (245) and then only updating the noise estimates during pauses in an information signal, when speech or other information is not detected, rather than continuously updating the noise estimates. A noise estimate is calculated for each frequency band and provides for the inclusion of a variable mathematical factor that can be set by the user to produce the best sound quality.

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1 **A METHOD AND SYSTEM FOR UPDATING NOISE ESTIMATES DURING**
2 **PAUSES IN AN INFORMATION SIGNAL**

3
4 **FIELD OF THE INVENTION**

5
6 This invention relates to a method and system for improving Adaptive Speech
7 Filter (ASF) estimates of the noise component of complex signals that contain both the
8 information signal and noise. The present invention generates noise estimates that are
9 updated only during pauses of the information signal. This produces an increase in
10 processing speed and a decrease in system memory. The methods of the present
11 invention are particularly suited to implementation on inexpensive digital signal
12 processors.

13
14 **BACKGROUND OF THE INVENTION**

15
16 The spectral components of an information signal are used in a number of signal
17 processing systems including channel vocoders for communication of speech, speech
18 recognition systems and signal enhancement filters. Since the inputs to these systems
19 are often contaminated by noise there has been a great deal of interest in noise reduction
20 techniques and consequently noise estimation techniques. The effect of uncorrelated
21 noise is to add a random component to the power in each frequency band, and the
22 subject of accurately assessing the noise content is crucial to achieve the desired end
23 result, which is the elimination of noise from the complex signal.

24
25 Noise-free spectral components are required for optimum operation of channel
26 vocoders. In a vocoder the input signal is filtered into a number of different frequency
27 bands and the signal from each band is rectified (squared) and smoothed (low pass
28 filtered). The smoothing process tends to reduce the variance of the noise. Such
29 methods are disclosed in U.S. Patent No. 3,431,355 to Rothauser et al and U.S. Patent
30 No. 3,431,355 to Schroeder. An alternative approach is disclosed in U.S. Patent No.
31 3,855,423 to Brendzel et al. In this approach the level of the noise in each band is
32 estimated from successive minima of the energy in that band and the level of the signal

1 is estimated from successive maxima. In U.S. Patent No. 4,000,369 to Paul et al, the
2 noise levels are estimated in a similar fashion and subtracted from the input signals to
3 obtain a better estimate of the speech signal in each band. This method reduces the
4 mean value of the noise.

5
6 Another application of spectral processing is for speech filtering. Weiss et al., in
7 "Processing Speech Signals to Attenuate Interference", presented at the IEEE Symp.
8 Speech Recognition. April 1974, disclose a spectral shaping technique. This technique
9 uses frequency domain processing and describes two approaches - amplitude modulation
10 (which is equivalent to gain control) and amplitude clipping (which is equivalent to a
11 technique called spectral subtraction). Neither the noise estimate nor the speech
12 estimate is updated so this filter is not adaptive. An output time waveform is obtained
13 by recombining the spectral estimates with the original phases.

14
15 An adaptive speech filter is disclosed in U.S. Patent No 4,185,168 to Graupe and
16 Causey, which is included by reference herein. Graupe and Causey describe a method
17 for the adaptive filtering of a noisy speech signal based on the assumption that the noise
18 has relatively stationary statistics compared to the speech signal.

19
20 In Graupe and Causey's method the input signal is divided into a set of signals
21 limited to different frequency bands. The signal to noise ratio for each signal is then
22 estimated in accordance with the time-wise variations of it's absolute value. The gain of
23 each signal is then controlled according to an estimate of the signal to noise ratio (the
24 gain typically being close to unity for high signal to noise ratio and less than unity for
25 low signal to noise ratio).

26
27 Graupe and Causey describe a particular method for estimating the noise power
28 from successive minima in the signals, and describe several methods for determining the
29 gain as a function of the estimated noise and signal powers. This is an alternative to the
30 method described earlier in U.S. Patent No. 4,025,721 to Graupe and Causey, which
31 detects the pauses between utterances in the input speech signal and updates estimates of
32 the noise parameters during these pauses. In U.S. Patent No. 4,025,721, Graupe and

1 Causey describe the use of Wiener and Kalman filters to reduce the noise. These filters
2 can be implemented in the time domain or the frequency domain.

3
4 Boll, in "Suppression of Acoustic Noise in Speech using Spectral Subtraction",
5 IEEE Transactions on Acoustics, Speech and Signal Processing, Vol. ASSP-27, No. 2,
6 April, 1979, describes a computationally more efficient way of doing spectral
7 subtraction. In the spectral subtraction technique, used by Paul, Weiss and Boll, a
8 constant or slowly varying estimate of the noise spectrum is subtracted. However,
9 successive measurements of the noise power in each frequency bin vary rapidly and only
10 the mean level of the noise is reduced by spectral subtraction. The residual noise will
11 depend upon the variance of the noise power. This is true also of Weiss's spectral
12 shaping technique where the spectral gains are constant. In Graupe's method the gain
13 applied to each bin is continuously varied so that both the variance and the mean level of
14 the noise can be reduced.

15
16 There are many schemes for determining the spectral gains. One scheme is
17 described by Ephraim and Malah in "Speech enhancement using a minimum mean-
18 square error short-time spectral amplitude estimator", IEEE Transactions on Acoustics,
19 Speech and Signal Processing, Vol. ASSP-32, No. 6, Dec. 1984. This describes a
20 technique for obtaining two estimates of the signal to noise ratio - one from the input
21 signal and one from the output signal. It does not update the estimate of the noise level.
22 The gain is a complicated mathematical function of these two estimates, so this method
23 is not suitable for direct implementation on a digital processor.

24
25 In U.S. Patent No. 5,012,519 to Aldersburg et al the gain estimation technique of
26 Ephraim and Malah is combined with the noise parameter estimation method disclosed
27 in U.S. Patent No. 4,025,721 to Graupe and Causey to provide a fully adaptive system.
28 The mathematical function of Ephraim and Malah is replaced with a two-dimensional
29 lookup table to determine the gains. However, since the estimates of the signal to noise
30 ratio can vary over a very large range, this table requires a large amount of expensive
31 processor time and memory. Aldersburg et al use a separate voice detection system on
32 the input signal which requires significant additional processing time.

There is thus an unmet need in the art to be able to utilize an efficient adaptive signal processing technique for the accurate and fast identification of noise. Processing time and memory efficiency would be improved if the noise estimates were only done during pauses of the information signal, so that noise estimates are updated only when an information signal is not detected. The algorithm should be capable of being implemented on inexpensive digital signal processors.

SUMMARY OF THE INVENTION

It is an object of the present invention to be able to obtain and update noise estimates only during pauses of the information signal, thereby decreasing processing time and memory requirements.

Therefore, according to the present invention, a method and system provide for noise estimates to be updated only during pauses in an information signal, when speech or other information is not detected, rather than continuously updating the noise estimates. Waiting for pauses in the information signal before updating the noise estimates allows processing time and memory requirements to be decreased. It also allows adaptive speech filtering to be easily implemented on inexpensive digital signal processors.

According to the method of the present invention, after a set of input frequency components have been produced, the total power calculated for each input frequency component, the power of the information signal estimated, a modified gain of the information signal calculated, and the input frequency component multiplied by the modified gain to produce an estimate of the power of the frequency component, then an estimate of the noise power is updated only if a pause in the information signal has been detected. Detecting a pause in the information signal is accomplished by first determining whether the estimate of the power of the frequency component of the information signal exceeds a first predetermined threshold value at each frequency. If the estimate of the power does exceed the first predetermined threshold value, then a threshold value $\text{thrshldCnt}[f]$ is checked to determine if it exceeds a second

1 predetermined threshold value. If the threshold value does exceed the second
2 predetermined threshold value, then a pause has been detected. If the threshold value
3 does not exceed the second predetermined threshold value, then no pause has been
4 detected. In this instance, the noise estimate is not updated and instead the threshold
5 value $\text{thrshldCnt}[f]$ is incremented.

6
7 The foregoing method of the present invention is implemented by a system for
8 estimating the noise power of frequency components of an information signal from an
9 input signal containing both the information signal and noise. The system has means to
10 produce input frequency components, one frequency component for each frequency
11 band, a first calculating means for calculating the total power of each input frequency
12 component, a second calculating means for calculating the modified gain of each
13 frequency band, and a gain multiplying means for multiplying the input frequency
14 component by the gain to produce an estimate of the power of the frequency component
15 of the information signal. The system additionally has an estimating means that
16 estimates the power of the information signal and updates the estimate of the noise
17 power only during a pause detected in the information signal by the estimating means.
18 The estimating means itself has an adder, a first comparison element coupled to the
19 adder that receives the estimate of the power of the information signal and a first
20 predetermined threshold value from the adder and determines whether the estimate of
21 the power of the information signal exceeds the first predetermined threshold value, and
22 a second comparison element coupled to the first comparison element that determines
23 whether a threshold value exceeds a second predetermined threshold value if the
24 estimate of the power of the information signal exceeds the first predetermined threshold
25 value. The estimate of the noise power is updated if the threshold value exceeds the
26 second predetermined threshold value since this condition is indicative of a pause
27 detected in the information signal. If a pause is not detected then an increment element
28 increments the threshold value.

29
30
31

BRIEF DESCRIPTION OF THE DRAWINGS

The novel features believed characteristic of the invention are set forth in the appended claims. The invention itself, however, as well as a preferred mode of use, and further objects and advantages thereof, will best be understood by reference to the following detailed description of an illustrative embodiment when read in conjunction with the accompanying drawing(s), wherein:

Figure 1 is a block diagram of a prior art system;

Figure 2 is a block diagram of a typical system of the present invention;

Figure 3 is a block diagram of a sub-system for gain modification, according to the present invention;

Figure 4 is a block diagram of a sub-system for signal power estimation, according to the present invention;

Figure 5 is a block diagram of a sub-system for noise power estimation, according to the present invention; and

Figure 6 is a block diagram of a sub-system for an information signal detector, according to the present invention.

DESCRIPTION OF THE INVENTION

The present invention describes a method for generating noise estimates which are only updated when an information signal, such as speech, is not detected. The noise estimate is calculated for each frequency band, and provides for the inclusion of a variable mathematical factor that can be set by the user in order to produce the best sound quality. The noise estimate method of the present invention allows the adaptive speech filter algorithm to perform better under all conditions.

The adaptive speech filtering of the present invention is a modified version of that described in U.S. Patent No. 4,185,168 to Graupe and Causey which describes a method for the adaptive filtering of a noisy speech signal. The method is based on the assumption that the noise has relatively stationary statistics compared to the speech signal.

The input to the filter is usually a digital signal obtained by passing an analog signal, containing noise and the information signal, through high- and low-pass filters and then sampling the resulting signal at a sample rate of at least 8 kHz. The high pass filter is designed to remove low frequency noise that might adversely affect the dynamic range of the filter. The turnover frequency of the high pass filter is less than f_{low} , where f_{low} is the lower limit of the speech band in Hertz. The low pass filter is an anti-aliasing filter, which has a turnover frequency of at least f_{high} , where f_{high} is the upper limit of the speech band in Hertz. The order of the low pass filter is determined by the sampling frequency and the need to prevent aliasing.

The output signal is calculated by filtering the input signal using a frequency domain filter with real coefficients and may be a time series or a set of spectral estimates. If the output is a time series then it may be passed to a digital to analog converter (DAC) and an analog anti-imaging filter to produce an analog output signal or it may be used as an input to subsequent signal processing.

The estimator of the spectral components comprises four basic steps:

1. Calculation of the spectrum of the input signal.

1 2. Estimation of the signal and noise power in each frequency bin within the
2 speech band ($f_{low} \rightarrow f_{high}$ Hz).

3 3. Calculation of the gains (coefficients) of the frequency domain filter for each
4 frequency bin, and

5 4. Calculation of the spectral estimates by multiplying each input spectral
6 component by the corresponding gain.

7 This is basically the method of Graupe and Causey, and each of the processes is
8 discussed below.

9
10 The estimates of the noise are updated during pauses in the information signal.
11 These pauses are detected by looking at the power estimate to see if it exceeds a
12 predetermined threshold, noise threshold, multiplied by noise[f] at each frequency. If
13 the power estimate is above the calculated threshold then a thrshldCnt[f] is checked to
14 see if it exceeds a predetermined value update_delay.

15
16 The spectral components of the input signal can be obtained by a variety of
17 means, including band pass filtering and Fourier transformation. In one approach a
18 discrete or fast Fourier transform is used to transform sequential blocks of N points of
19 the input time series. A window function, such as a Hanning window, can be applied, in
20 which case an overlap of $N/2$ points can be used. A Discrete Fourier Transform (DFT)
21 can be used at each frequency bin in the speech band or, alternatively, a Fast Fourier
22 Transform (FFT) can be used over the whole frequency band. The spectrum is stored
23 for each frequency bin within the speech band. For some applications it is desirable to
24 have unequally spaced frequencies - in these applications a Fast Fourier transform
25 cannot be used and each component may have to be calculated independently. In one
26 approach the input spectrum, X , is calculated as the Fourier transform of the input time
27 series, x , namely

28
$$X = \text{Fourier transform} \{ x, \text{window function}, N \}.$$

29
30 The power in the input spectrum is given by

31
$$\text{power} = \text{modulus squared}\{X\}.$$

Alternatively, a band pass filter may be used, in which case the power may be estimated by rectifying and smoothing the filter output. This version of a Graupe and Causey system is shown in Figure 1, Block Diagram 100. Input Time Signal 105, x , is applied to a bank of band pass filters. One of these bandpass filters is represented by Bandpass Filter 110 in Figure 1. The output of Bandpass Filter 110 is Input Spectral Signal 115, referred to as X . The power of Input Spectral Signal 115 is measured by Input Spectral Signal Power Measurement 140, which generates Total Input Spectral Power Signal 165. The method requires that estimates be made for both Total Input Spectral Power Signal 165 and Noise Power Estimator Output 160. Noise Power Estimator Output 160 is generated by Noise Power Estimator 145 which utilizes a time constant related to the time over which the noise content of Total Input Spectral Power Signal 165 can be considered stationary. Total Input Spectral Power Signal 165 is estimated by Signal Power Estimator 155. From these estimates Wiener Gain Coefficients 170 is calculated by Wiener Gain Calculator 150, Wiener Gain Calculator 150 determines the ratio of the power in the information signal, which is Total Input Spectral Power Signal 165, to the total power which is the sum of Noise Power Estimator Output 160 and Total Input Spectral Power Signal 165. For each frequency bin this is

$$W = \text{signal} / (\text{noise} + \text{signal}).$$

In the method of Graupe and Causey the Wiener gain, W , is directly applied to the corresponding component of the input spectrum. In the unmodified scheme the spectral components of the output are given by multiplying Input Spectral Signal 115 by Wiener Gain Coefficients 170 in Multiplier 120. The result is

$$Y = W * X$$

which is Output Spectral Signal 125. If Output Time Signal 135, y , is required it can be calculated by an inverse FFT (or DFT) and the 'overlap-add' method or by summing the components from individual channels using Channel Combiner 130.

After each iteration k the output block of N time points is updated as

$$y_k(1:N) = \text{inverse Fourier transform } \{Y, N\}$$

$$y_k(1:N/2) = y_k(1:N/2) + y_{k-1}(N/2+1:N)$$

1 The first $N/2$ points of y_k are then sent to Channel Combiner 130 or may be used for
 2 further processing.

3
 4 An improved system is shown in Figure 2, Block Diagram 200. The additional
 5 features are described below.

6 7 **Gain Modification**

8 Time Input Signal 205 is applied to Bandpass Filter 210. The output of
 9 Bandpass Filter 210 is applied to the input of Multiplier 220, and if a time signal output
 10 is desired Channel Combiner 230 is utilized to generate Time Output Signal 235. When
 11 the signal to noise ratio is low the direct use of the Wiener gain results in a residual
 12 noise which has a musical or artificial character. One improvement is the use of Gain
 13 Modifier 270, which reduces the musical nature of the residual noise. Gain Modifier
 14 270 receives inputs from Wiener Gain Calculation 250 and Noise Power Estimator 245.
 15 The output of Total Input Spectral Power Measurement 240 is also routed as an input to
 16 Gain Modifier 270.

17
 18 Gain Modifier 270 is presented by Figure 3, Block Diagram 300. The
 19 instantaneous power of the information signal can be estimated as the product of the
 20 instantaneous power and the Wiener gain. This gives an estimate of the instantaneous
 21 signal to noise ratio, snr , in each frequency bin obtained by dividing Total Input
 22 Spectral Power 265 by Noise Power Estimator Output 260, which is accomplished by
 23 Divider 305, and using this quotient to modulate or multiply Wiener Gain Coefficients
 24 280. This is accomplished by Multiplier 325, and the output of Multiplier 325 is Signal-
 25 to-Noise Ratio Estimate 320. Hence

$$26 \quad snr = W * (power / noise).$$

27 A function of the signal to noise ratio is then calculated by Function Modifier 315, and
 28 Modified Coefficients 275, which are denoted by the vector C , are calculated by
 29 dividing the output of Function Modifier 315 by the output of Divider 305. This is
 30 accomplished by Divider 310 and is done for each frequency, so that

$$31 \quad C = F\{snr\} * (noise / power) = F\{snr\} / (power / noise)$$

where F is a function of a single variable and is therefore well suited to implementation on a DSP as a look-up table or an analytic function. One form of the function F is given by

$$F(x) = \begin{cases} x^{1/2}, & x < \text{snr0} \\ x + c, & x \geq \text{snr0} \end{cases}$$

where c and snr0 are constants. Other forms can be used, but it is desirable that the function is approximately linear at high signal to noise ratios. In particular the gain of Ephraim and Malah may be manipulated so that it can be implemented in this form.

Output Spectral Signal 225, Y , which is the estimate of the spectrum of the information signal, is calculated by multiplying 215 by the corresponding Modified Coefficients 275, as shown in Figure 2, so that for each frequency

$$Y = C * X$$

Signal Estimation

Ephraim and Malah describe a method for updating a signal to noise ratio. This method can be modified to give an estimate of Signal Power Estimator Output 285. Signal Power Estimator 255 uses the power in the output spectral signal Output Spectral Signal Power 290 which is calculated by Output Spectral Signal Power Measurement 295 as shown in Figure 2. The method is shown in detail in Figure 4. Block Diagram 400, and is given by

$$\begin{aligned} \text{sig1} &= \text{maximum}\{\text{power} - \text{noise}, 0\} \\ \text{sig2} &= \text{modulus squared}\{Y\} \\ \text{signal} &= (1 - \text{beta}) * \text{sig1} + \text{beta} * \text{sig2} \end{aligned}$$

The difference between Total Input Spectral Power 265 and Noise Power Estimator Output 260 is calculated by Adder 405. The output of Adder 405 is half-wave rectified by Half-wave Rectifier 410. The output of Half-wave Rectifier 410 is Half-wave Rectifier Output Signal 415, and Half-wave Rectifier Output Signal 415 is weighted by (1-Beta) Weighting Function 420. Signal Power Estimator Output 285 is obtained as the sum of the output of (1-Beta) Weighting Function 420 and the output of (Beta) Weighting Function 430 by Adder 425. The output of (Beta) Weighting Function

430 is a weighted value of Output Spectral Signal Power 290. The weighting parameter *beta* used in the weighted sum is typically chosen to be greater than 0.9 and less than 1.

Noise Estimation

The estimates of the noise can be updated during the pauses in the information signal. The pauses can be detected by looking at the power estimate to see if it exceeds a predetermined threshold, noise threshold multiplied by noise[f] at each frequency. If the power estimate is above the calculated threshold then a thrsholdCnt[f] is checked to see if it exceeds a predetermined value update_delay. If it does, the noise estimate is updated as

$$\text{Noise}[f] = \alpha * \text{power} + (1 - \alpha) * \text{noise}[f]$$

$$\text{Noise}[f] = \text{maximum} \{ \text{noise}[f], \text{minNoise} \}$$

$$\text{ThrsholdCnt}[f] = 0$$

MinNoise is a constant that prevents noise[f] from being equal to zero. It is typically equal to $1 * 10 \exp -7$.

If a pause is not detected, thrsholdCnt[f] is incremented

$$\text{ThrsholdCnt}[f] = \text{thrsholdCnt}[f] + 1$$

This type of noise estimator is depicted in Figure 5, Block Diagram 500. Input Spectral Power 265 is applied to a first input of Adder 575. Noise Power Estimator Output 260 is applied to the input of Time Delay 565. Time Delay 565 functions as a one-sample delay. The output of Time Delay 565 is multiplied by a constant, Noise Threshold, in Noise Threshold Multiplier 570. The output of Noise Threshold Multiplier 570 is routed to a second input of Adder 575. The output of Adder 575 is input to (≥ 0 ?) Function 550. The inputs of the logical AND 556 enables Algorithmic Process 545 only if Function 550 and Function 555 are true. If the output of function 556 is False Algorithmic Process 545 is disabled. The output of Logical AND 556 is applied to the input of Inverter 557. A false input at Inverter 557 will enable Algorithmic Process 540.

The sequence of First Algorithmic Process 540 will now be described. Input Spectral Power 265 is input to (alpha) Multiplier 505. The output of (alpha) Multiplier

505 is applied to a first input of Adder 510. The output of Adder 510 is an input to Multiplier 525. Time Delay 520 is a one sample delay. The output of Multiplier 525 is Noise Power Estimator Output 260. Noise Power Estimator Output 260 is applied to the input of Time Delay 565 and to the input of Time Delay 530. Time Delay 530 functions as a single sample delay. The output of Time Delay 530 is applied to the input of (1-alpha) Multiplier 515. The output of (1-alpha) Multiplier 515 is applied as a second input of Adder 510.

Second Algorithmic Process 545 produces an increment in the value of ThrsholdCnt, and is represented by (Increment thrsholdCnt) Function 535.

Information Signal Detector

The present invention operates to update estimates of the noise during pauses in the information signal. The presence of an information signal can be detected by looking at a weighted sum of the signal to noise components across frequency bins (a uniform weighting may be used). If this weighted sum is above a predetermined threshold, the signal is assumed to contain information and the noise estimate is updated. This is shown in Figure 6, Block Diagram 600. Signal-to-Noise Ratio Estimates 605 are weighted by Signal-to-Noise Ratio Weighting Coefficients 610 and then summed by Summer 615 to produce Summer Output Signal 630. S, before being input to Threshold Detector 620. The output of Threshold Detector 620 is Threshold Detector Output Signal 625.

One algorithmic example is described below:

at each update number k

$X = \text{Fourier transform} \{ x, \text{window function}, N \}.$

FOR each frequency number f in speech band

$power = \text{modulus squared} \{ X[f] \}$

$sig1 = \text{maximum} \{ power - noise[f], 0 \}$

$sig2 = \text{modulus squared} \{ Y[f] \}$

$signal = (1 - \beta) * sig1 + \beta * sig2$

$W = signal / (noise[f] + signal)$

14

```

1       $snr = W * (power / noise[f])$ 
2       $C = F\{snr\} / (power / noise[f])$ 
3      IF(power-noiseThreshold*noise[f]>=0 and thrsholdCnt<update_delay
4          THEN thrsholdCnt[f]=thrsholdCnt[f]+1
5          OTHERWISE noise[f]=alpha*power+(1-alpha)*noise[f]
6              Noise[f]=max(noise[f], minNoise)
7              ThrsholdCnt[f]=0
8          ENDIF
9       $old\_power[f] = power$ 
10      $Y[f] = C * X[f]$ 
11     ENDFOR
12      $y_k(1:N) = \text{inverse Fourier transform } \{Y, N\}$ 
13      $y_k(1:N/2) = y_k(1:N/2) + y_{k-1}(N/2+1:N)$ 
14     At the end of each iteration,  $k$ , the signal  $y_k(1:N/2)$  provides an estimate of the
15     information signal. If a pause is not detected in the information signal, then
16     thrsholdCnt[f] is incremented:  $thrsholdCnt[f] = thrsholdCnt[f] + 1$ .

```

17

18 As can be seen from the foregoing description the present invention teaches a
19 method whereby a noise estimate may be calculated and utilized in an adaptive speech
20 filter algorithm. The noise estimation method generates noise estimates only during
21 pauses in the information signal, rather than continuously updating the noise estimates.
22 This noise estimation and updating technique allows for faster convergence and quicker
23 cancellation of interfering tones than prior art techniques. The algorithmic technique
24 can be implemented on inexpensive digital signal processors. It typically will result in
25 less processing time, and memory requirements are less. The method of the present
26 invention avoids corruption of the noise estimates due to additive information signal
27 content that is common in other methods of noise estimation.

28

29 While the invention has been particularly shown and described with reference to
30 a preferred embodiment, it will be understood by those skilled in the art that various
31 changes in form and detail may be made therein without departing from the spirit and
32 scope of the invention.

1 What is claimed is:

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1. A method for estimating the frequency components of an information signal from an input signal containing both the information signal and noise, said method comprising:

 producing a set of input frequency components, one for each frequency band and for each frequency component;

 calculating the total power in each input frequency component;

 estimating the power of the information signal included therein;

 calculating a modified gain for each frequency band as a function of the total power, the estimate of the power of the information signal and an estimate of the noise power;

 multiplying the input frequency component by said gain to thereby produce an estimate of the power of the frequency component of said information signal;

 detecting a pause in the information signal, further comprising the following steps:

 determining whether the estimate of the power of the frequency component of said information signal exceeds a first predetermined threshold, and

 if the estimate of the power of the frequency component of said information signal exceeds the first predetermined threshold, then determining whether a threshold value exceeds a second predetermined threshold wherein if the pause is detected if the threshold value exceeds the second predetermined threshold, and

1 updating the estimate of the noise power during the pause detected in the
2 information signal.

3
4 2. A method as in claim 1 wherein if the threshold value does not exceed the
5 second predetermined threshold, then incrementing the threshold value.

6
7 3. A method as in claim 1 and including the step of estimating the overall signal to
8 noise ratio from a weighted sum of the estimated signal to noise ratios of each frequency
9 band.

10
11 4. A method as in claim 3 and including the step of using said estimated overall
12 signal to noise ratio to determine the presence of an information signal in the input
13 signal.

14
15 5. A method as in claim 1 and including the step of recombining the estimates of
16 the frequency components of said information signal to produce a noise reduced output
17 signal.

18
19 6. A method as in claim 1 in which the power of the information signal is estimated
20 from a combination of a previous estimate of the frequency components of said
21 information signal and the positive difference between the power in the input frequency
22 component and the estimate of the noise power.

23
24 7. A method as in claim 6, in which the gain in each frequency band is determined
25 by the steps of:

26
27 estimating a Wiener gain from said estimate of the noise power and the estimate
28 of the power of the information signal;

29
30 multiplying said Wiener gain by the ratio of the power of the input frequency
31 component to the estimated noise power to produce an estimate of the signal to noise
32 ratio;

1
2 calculating a function of the estimated signal to noise ratio; and

3
4 dividing said function of the estimated signal to noise ratio by the ratio of the
5 power of the input frequency component to the estimated noise power to thereby
6 produce a modified gain.

7
8 8. A method as in claim 1 in which the filtering is performed via a Fourier
9 transform.

10
11 9. A method as in claim 1 which is used as a preprocessor to a speech or voice
12 recognition system.

13
14 10. A method as in claim 1 which is used for reducing noise in a communications
15 system.

16
17 11. The method of claim 1 wherein said means to produce a set of input frequency
18 components comprises filtering the input signal through a set of band pass filters.

19
20 12. The method of claim 1 wherein said means to produce a set of input frequency
21 components comprises calculating the Fourier Transform of the input signal.

22
23 13. A system for estimating the noise power of frequency components of an
24 information signal from an input signal containing both the information signal and
25 noise, said system comprising:

26
27 means to produce a set of input frequency components, one for each frequency
28 band and for each frequency component;

29
30 a first calculating means for calculating the total power in each input frequency
31 component;

32

1 an estimating means for estimating the power of the information signal included
2 therein and for updating an estimate of the noise power only during a pause detected in
3 the information signal by the estimating means;

4
5 a second calculating means for calculating a modified gain for each frequency
6 band as a function of the total power, the estimate of the power of the information signal
7 and the estimate of the noise power; and

8
9 gain multiplying means for multiplying the input frequency component by said
10 gain to thereby produce an estimate of the power of the frequency component of said
11 information signal.

12
13 14. A system as in claim 13 wherein the estimating means comprises:

14
15 an adder that is provided with an input spectral power signal and a first
16 predetermined threshold value;

17
18 a first comparison element that receives the estimate of the power of the
19 information signal and the first predetermined threshold value from the first adder,
20 wherein the first comparison element determines whether the estimate of the power of
21 the information signal exceeds the first predetermined threshold value; and

22
23 a second comparison element coupled to the first comparison element that
24 determines whether a threshold value exceeds a second predetermined threshold value if
25 the estimate of the power of the information signal exceeds the first predetermined
26 threshold value, wherein if the threshold value exceeds the second predetermined
27 threshold value then the pause is detected and the estimating means updates the estimate
28 of the noise power.

29
30
31
32

1 15. A system as in claim 13 further comprising:

2
3 an increment element that increments the threshold value if the threshold value
4 does not exceed the second predetermined threshold value.

5
6 16. A system as in claim 13 in which the second calculating means comprises:

7
8 means for estimating a Wiener gain from said estimate of the noise power and
9 the estimate of the power of the information signal;

10
11 Wiener multiplying means for multiplying said Wiener gain by the ratio of the
12 power of the input frequency component to the estimate of the noise power to produce
13 an estimate of the signal to noise ratio;

14
15 function calculating means for calculating a function of the estimated signal to
16 noise ratio; and

17
18 division means for dividing said function of the estimated signal to noise ratio by
19 the ratio of the power of the input frequency component to the estimate of the noise
20 power to thereby produce a modified gain.

21
22 17. The system of claim 13 wherein said means to produce a set of input frequency
23 components is capable of filtering the input signal through a set of band pass filters.

24
25 18. The system of claim 13 wherein said means to produce a set of input frequency
26 components is capable of calculating the Fourier Transform of the input signal.

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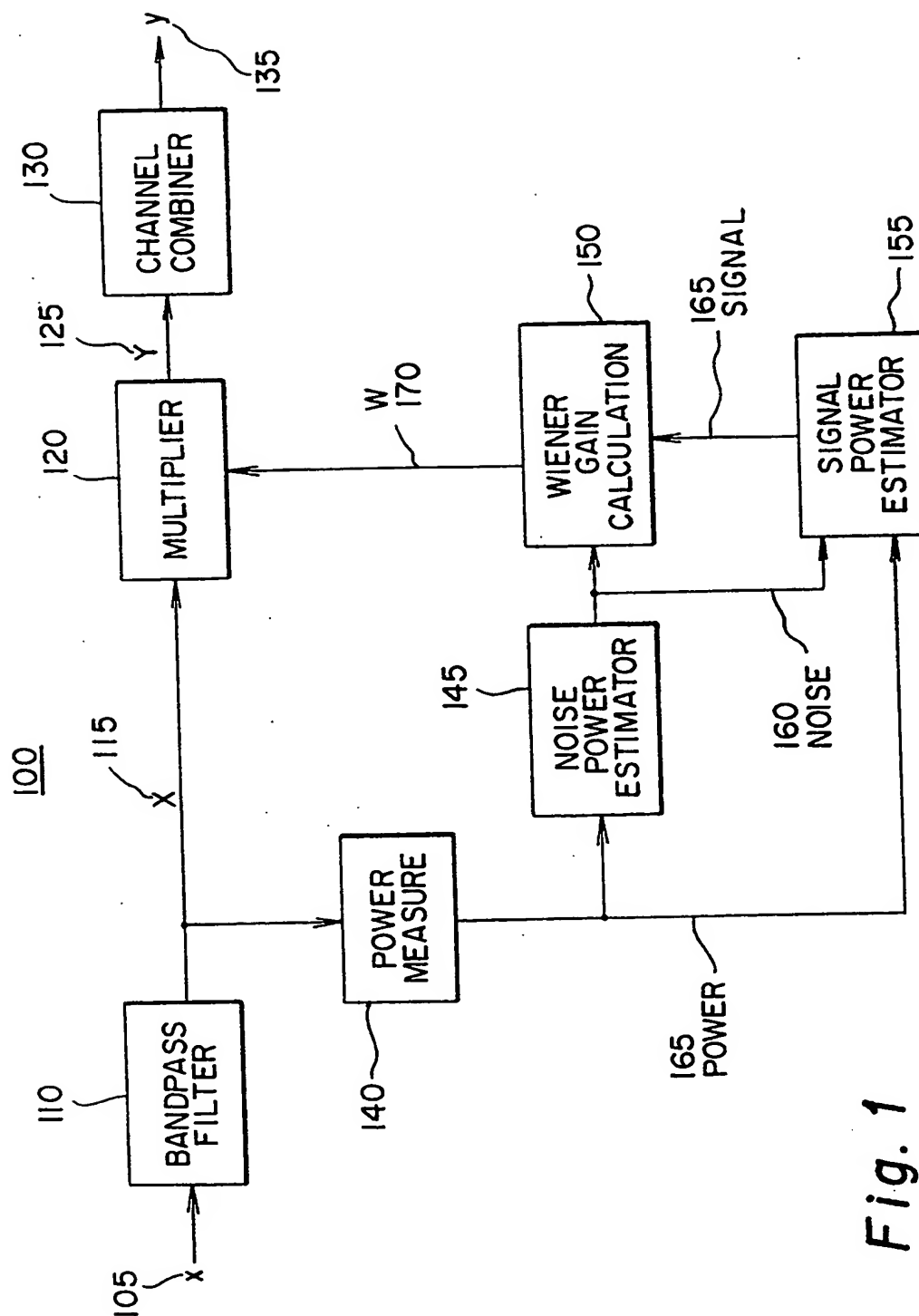


Fig. 1

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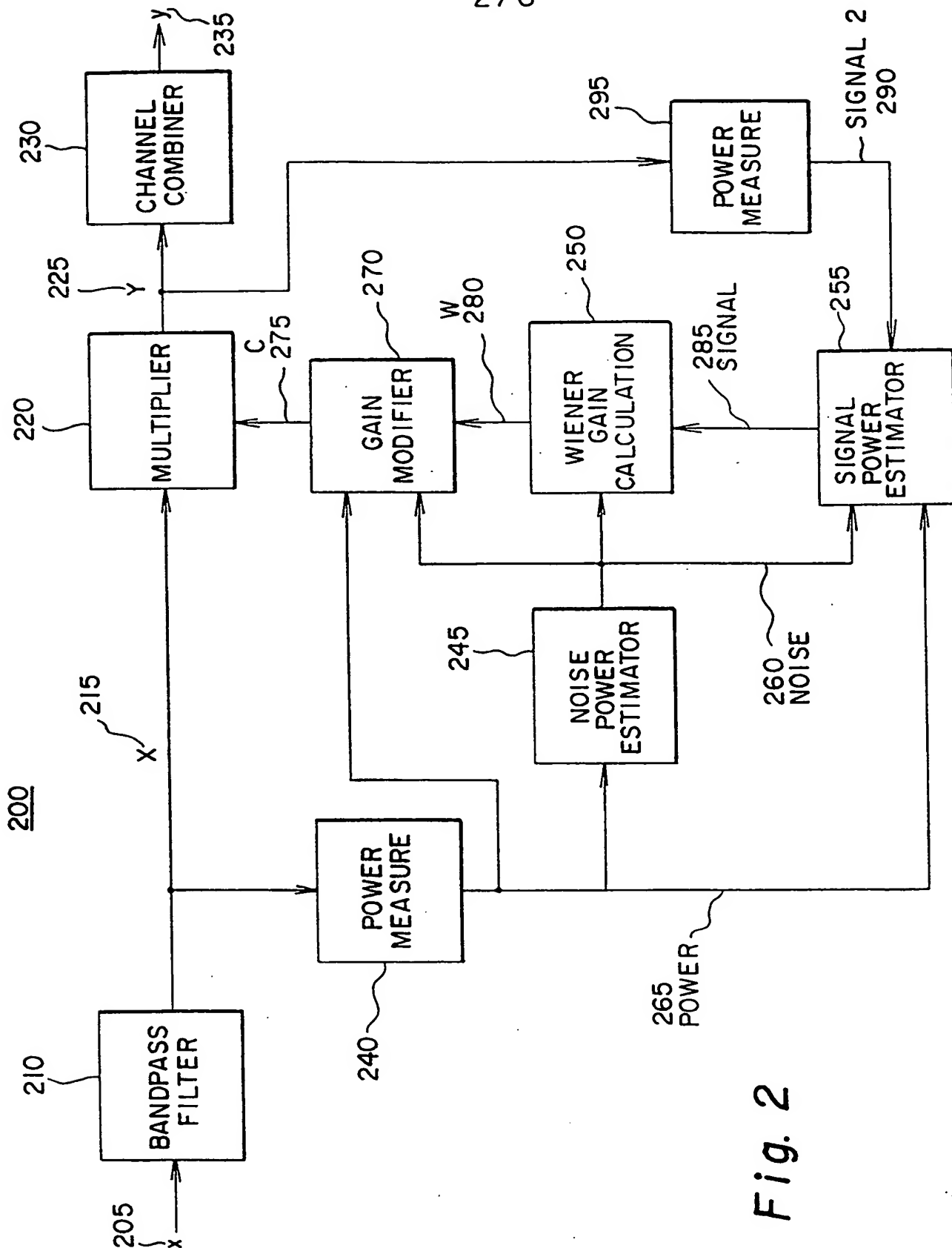


Fig. 2

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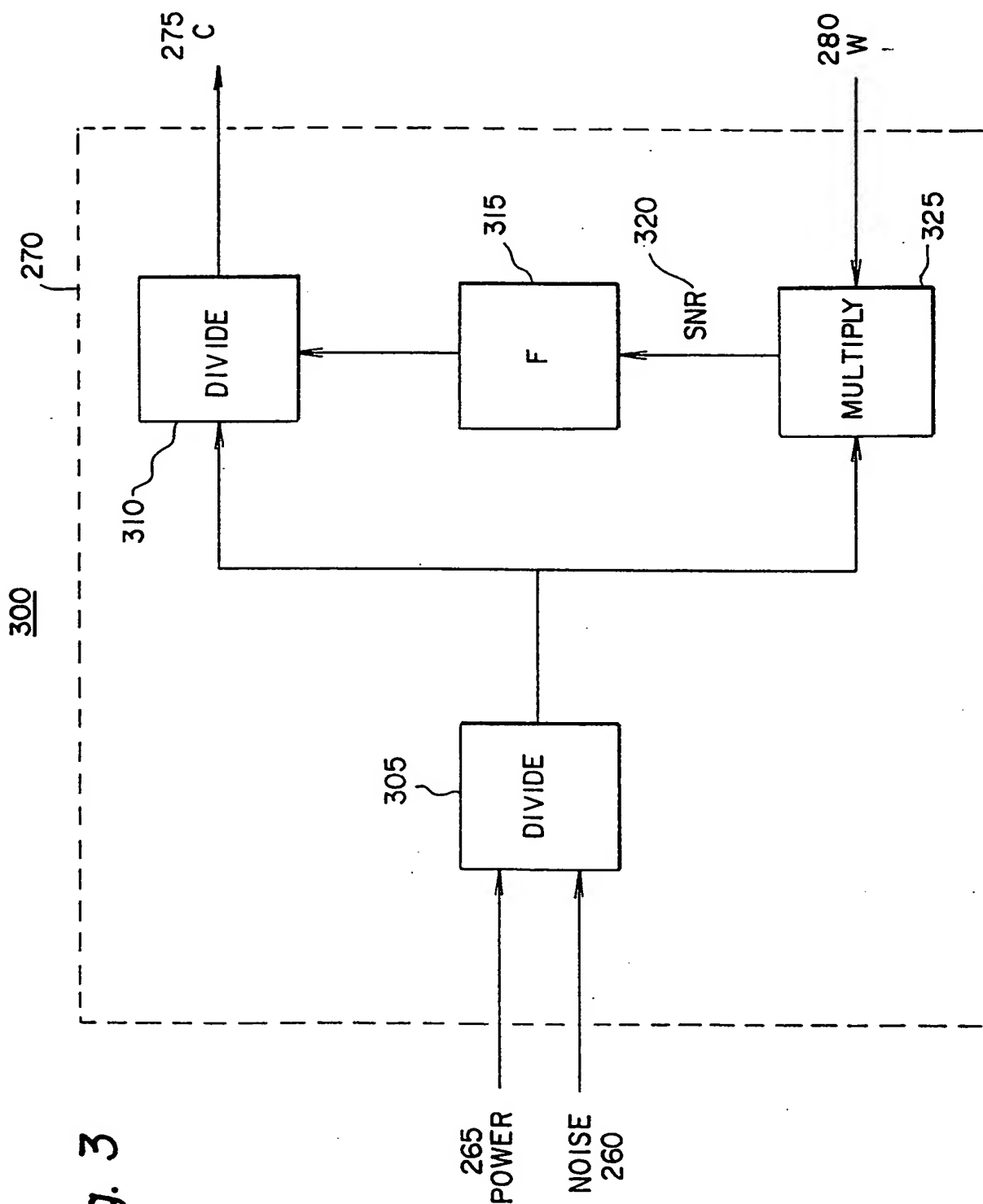


Fig. 3

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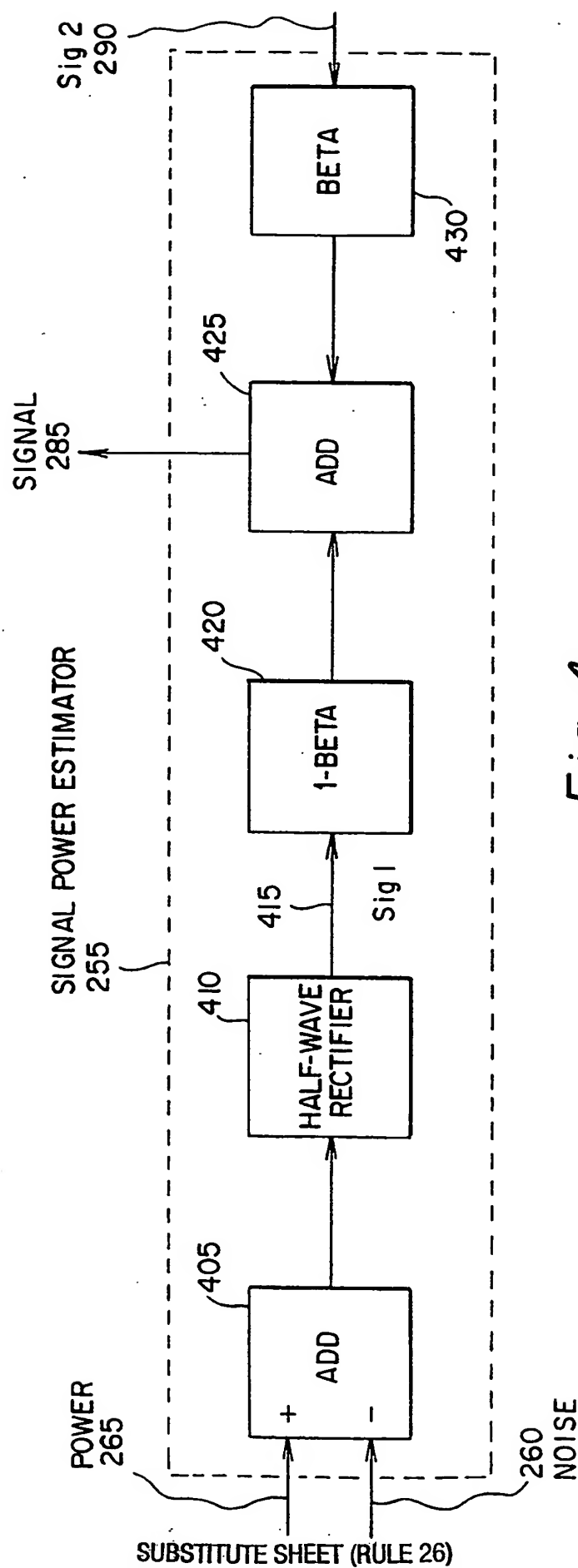
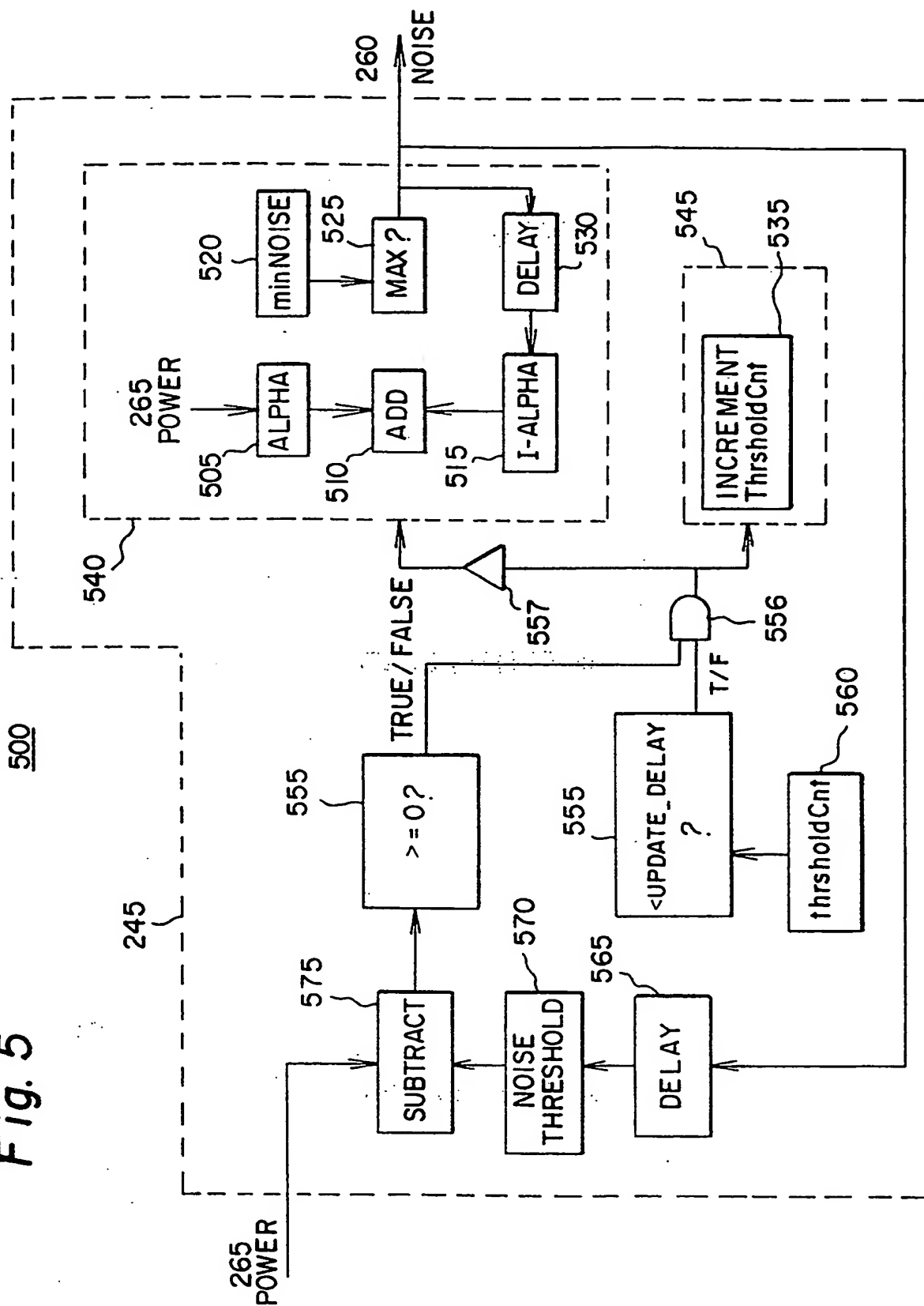


Fig. 4

Fig. 5



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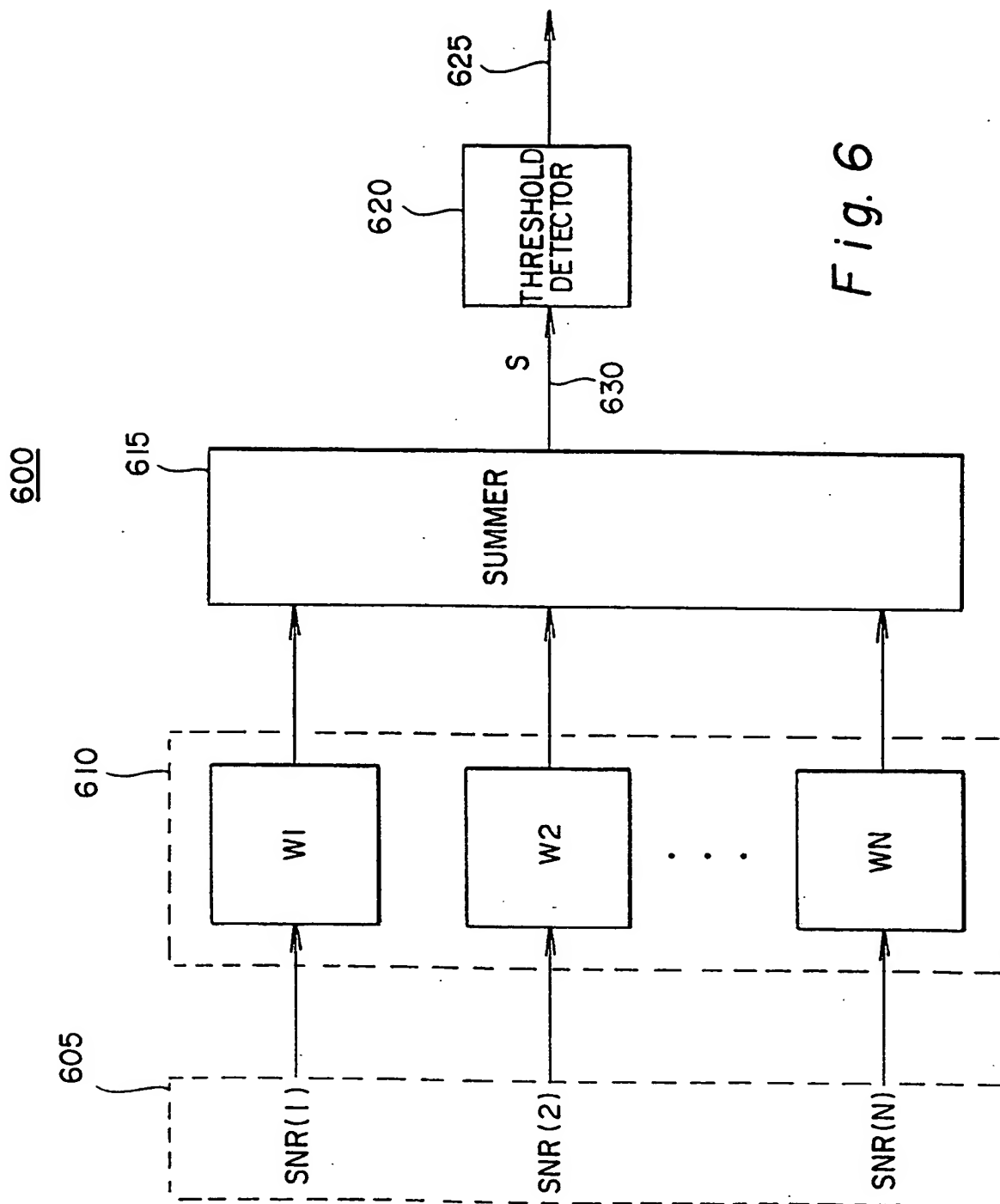


Fig. 6

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US99/23285

A. CLASSIFICATION OF SUBJECT MATTER

IPC(6) :G01R 23/16

US CL :702/75, 76, 77; 704/226, 205

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

U.S. : Please See Extra Sheet.

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched
None

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

USPTO APS West

search terms: pause, noise, estimate, estimates, estimation, power, signal, information

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
X	US 5,012,519 A (ADLERSBERG et al.) 30 April 1991 (30-04-1991), Abstract, col. 1, lines 21-29, col. 5, lines 28-50, col. 8, lines 55-68, col. 9, lines 19-40, col. 10, lines 18-22, 37-40.	1-18
Y	US 5,768,473 A (EATWELL et al) 16 June 1998 (16-06-1998), Abstract, col. 6, lines 57-67, col. 7, lines 1-27, col. 8, lines 49-54.	13, 16-18
Y	US 5,706,394 A (WYNN) 06 January 1998 (06-01-1998), Abstract, col. 2, lines 25-50, col. 9, lines 23-39.	13, 16-18



Further documents are listed in the continuation of Rev. C.



See patent family annex.

* Special categories of cited documents:	*T* later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
A document defining the general state of the art which is not considered to be of particular relevance	*X* document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
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L document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	*Z* document member of the same patent family
O document referring to an oral disclosure, use, exhibition or other means	
P document published prior to the international filing date but later than the priority date claimed	

Date of the actual completion of the international search

05 DECEMBER 1999

Date of mailing of the international search report

09 FEB 2000

Name and mailing address of the ISA/US
Commissioner of Patents and Trademarks
Box PCT
Washington, D.C. 20231

Facsimile No. (703) 305-3230

Authorized officer

for HAL D. WACHSMAN

Telephone No. (703) 305-5782

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US99/23285

B. FIELDS SEARCHED

Minimum documentation searched

Classification System: U.S.

702/75, 76, 77, 57, 60, 66-74, 79, 106, 111, 193, 124, 126, 183, 189-191, 195, 197, 198; 704/226, 205, 208, 210, 227-229, 225, 203, 233, 234; 381/317, 318, 320, 93, 98, 94.1-94.3; 708/322, 323, 309, 311; 324/76.19, 76.21, 76.22, 76.24, 613, 614; 375/232-234